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A Summary of the Present Position of Stereophonic Broadcasting

by

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BRITISH BROADCASTING CORPORATION

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BRITISH BROADCASTING CORPORATION

FOREWORD

THIS is one of a series of Engineering Monographs published by the British Broadcasting Corporation. About six are produced every year, each dealing with a technical subject within the field of television and sound broadcasting. Each Monograph describes work that has been done by the Engineering Division of the BBC and includes, where appropriate, a survey of earlier work on the same subject. From time to time the series may include selected reprints of articles by BBC authors that have appeared in technical journals. Papers dealing with general engineering developments in broadcasting may also be included occasionally.

This series should be of interest and value to engineers engaged in the fields of broadcasting and of telecommunications generally.

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A SUMMARY OF THE PRESENT POSITION OF STEREOPHONIC BROADCASTING

SYNOPSIS

The various methods by which stereophonic programmes can be produced for sound recording or broadcasting are discussed with particular reference to stereophonic reproduction under domestic conditions.

The problem of transmitting stereophonic programmes on existing radio-frequency channels, while providing for 'compatible' reception on ordinary broadcast receivers, is considered; the principles and potentialities of the main systems so far proposed are discussed. Attention is drawn to the difficulties which would arise in distributing stereophonic programmes to transmitters by line at audio frequency.

A brief reference is made to the possible application of stereophony to television sound.

1. Definitions

In a subject such as stereophony which has recently expanded rapidly it is not surprising that various expressions used are not clearly defined. To avoid ambiguity it is proposed, therefore, to preface this monograph with a brief explanation of some of the special terms employed.

Stereophony

The British Standards Institution, in its current *Glossary of Acoustical Terms,* defines a stereophonic sound system as

'a sound transmission system in which two or more channels are arranged to give to the listener an impression of the spatial distribution of the sounds'.

This description is open to the objection of being equally applicable to systems in which the listener receives no information about positions and for the purpose of this monograph the following definition will be used:

'Stereophony is the art or practice of employing two or more electro-acoustic transducers to give the listener an impression of the relative positions in space of a number of sources of sound'.

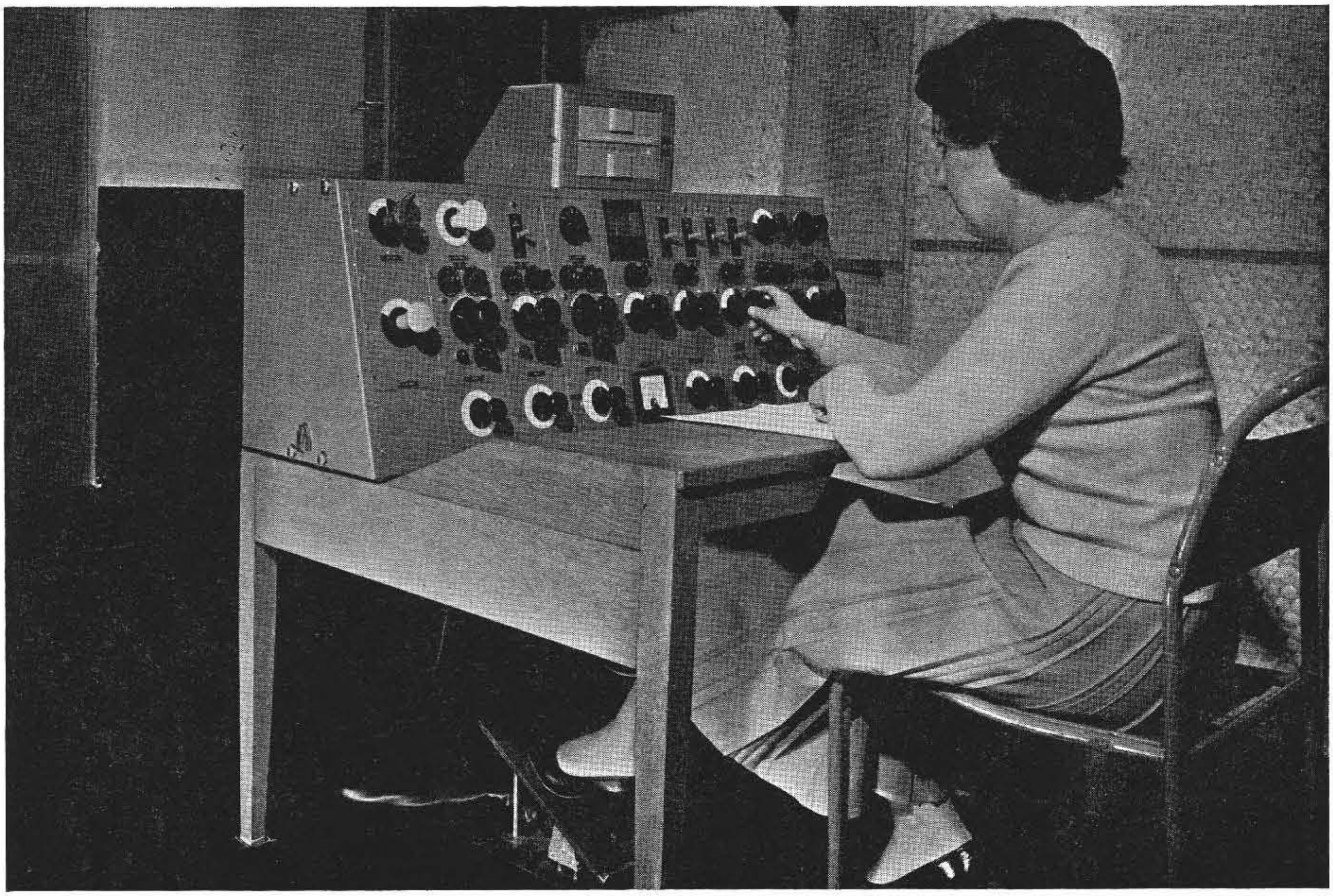

Fig. 1 — Experimental stereophonic control desk

*Monophony**

This term will be used to describe the present form of sound broadcasting and normal sound recording. The use of the term 'monaural' in this connection is to be deprecated, for even when a single loudspeaker is employed the listener uses both ears. The term 'single channel' can also be ambiguous because some systems of stereophony use only one transmission channel.

Compatibility

A stereophonic system is said to be 'compatible' when it can also yield a signal capable of giving satisfactory results on equipment designed for monophonic reproduction.

Sound Image

To avoid frequent repetition of such expressions as 'the point from which the reproduced sound appears to emanate', the term 'image' has been borrowed from optics; thus, each source of sound in the studio is said to produce a corresponding image in the listening room.

Sound Stage

The region, extending from the left-hand to the right-hand loudspeaker, in which the sound images appear. 'Sound field' is deprecated because the term 'field' has other connotations.

NOTE: In all domestic systems of stereophony as well as in the majority of cinema systems, the positional effects are confined to the horizontal plane. Fortuitous subjective effects of vertical displacement sometimes occur in stereophonic reproduction but these are not fully understood and are not completely under control.

Scale of Width

The relationship between the angular displacement of a source of sound and that of its reproduced image.

Spaced Microphones

A group of microphones producing stereophonic effects wholly or partly through differences in the times of arrival of a given sound.

Coincident Microphones

A pair of microphones mounted so close together that differences in the times of arrival of a given sound are negligible.

2. Introduction

The idea of a stereophonic sound reproduction is far from new. Lord Rayleigh,(1) in the 1896 edition of *The Theory of Sound*, described early investigations into binaural hearing, and in 1881 a system employing two microphones connected to several pairs of earphones was tried in the Paris Opera House.(2) Physiologists have long been interested in the mechanism of binaural hearing and investigations in this field are still going on; among these the work of Cherry and his associates at Imperial College is of particular interest. However, it was not until the development of loudspeakers that experiments in stereophony, as we now understand the term, were commenced.

The first large-scale experiment in stereophonic reproduction(3) was carried out by the Bell Laboratories in 1934. Signals from a number of microphones placed in a row in front of an orchestra were reproduced by a corresponding row of loudspeakers in front of the audience; this arrangement, which requires as many transmission channels as there are microphones or loudspeakers, has been described as the 'wave-front' system.

Most of the early stereophonic systems were developed in connection with the cinema. The techniques employed were based on the 'wave-front' system, though in more recent times it has become common to supplement the original microphone array by single microphones, the signal from which is divided by means of a 'panoramic potentiometer' or 'pan-pot' unequally between the loudspeakers; by this artifice a single image can be produced at any desired part of the stage or made to move about as required.

During the past three decades the possibility of domestic stereophonic reproduction has been considered by various workers in relation to both sound recording and broadcast transmission. For economic reasons the number of channels has to be kept to a minimum and is in practice assumed to be two. A 'wave-front' system of stereophony having only two channels gives unsatisfactory results since sound sources mid-way between the microphones sound too distant; in addition, it is difficult to achieve a smooth transition of the image from one side of the stage to the other when the source of sound moves. These difficulties, which result from the wide spacing between the left- and right-hand microphones, can be somewhat mitigated by the addition of a central microphone with its output equally divided between the two channels. Alternative systems have however been devised by Blumlein,(4,5) de Boer(6) and others, employing microphones which are coincident or are separated by a distance which is a small fraction of the spacing between the loudspeakers. A second difficulty which arises when only two channels are employed is that the position of a sound image varies considerably with the position of the listener, who, to receive the correct directional information, must be situated on a line running midway between the loudspeakers. Palliative measures, such as the use of loudspeakers of special directional characteristics, or of a third loudspeaker centrally placed and fed from both channels, have been suggested, but the fundamental problem remains unsolved. On the other hand there is an increasing body of evidence that the benefits of stereophonic reproduction are apparent to the majority of individuals even when listening under conditions which are far from ideal.

Although Blumlein had shown in a patent application(4) made as early as 1931 that domestic systems of stereophony using two channels were possible, no satisfactory means of recording and reproducing two synchronized signals were available at that time. Blumlein's patent

* *pace Wireless World*, Editorial, June 1959.

covered a method of recording two stereophonic signals on disk by utilizing the two walls of the groove but because of practical difficulties this was not at the time a commercial proposition.

During the immediate post-war period, the rapid development of the German system of magnetic recording on tape provided an ideal medium for recording a number of signals simultaneously and eventually opened the way for an attempt to provide domestic stereophonic reproduction on a commercial scale. A limited amount of recorded material of this kind has been generally available for some time, but the market for it has not yet been very extensive.

In the meantime, however, developments in disk recording and processing have at last made it possible to implement the original proposals of Blumlein in what is now known as the '45/45' system. Long-playing stereophonic disks are now in commercial production and general agreement has been reached throughout the recording industry regarding the standards to be adopted. The improvement in realism apparent even on quite inexpensive reproducing coupled with the unexpectedly low cost of the stereophonic disks suggests that, in the near future, interest in stereophony, if still far from universal, will be by no means confined to a small number of high quality enthusiasts. It would appear that gramophone recording is entering upon a new era, and a continuous extension in the field of domestic stereophonic reproduction is to be expected.

The possibility of distributing stereophonic programme material by broadcasting has been the subject of discussion from time to time and a few practical experiments have been carried out, but it has hitherto appeared unlikely that the public demand for such a service would ever be sufficient to justify the amount of engineering effort involved. This situation has been radically altered by the recent developments in the recording industry, and broadcasting organizations in Europe and the U.S. are now giving further consideration to the economic and engineering problems which would be involved in providing a stereophonic service. In the U.S., all the major firms in the communication world are working on the subject and experiments have been carried out on a number of systems. Early in 1959, a National Stereophonic Radio Committee* (N.S.R.C.) was set up to make recommendations to the Federal Communications Commission (F.C.C.) on transmission standards; the BBC is represented by an observer on one of the technical panels. The Comité Consultatif International des Radiocommunications (C.C.I.R.) has also accepted a study programme and a question on stereophonic broadcasting which follow closely the terms of reference of the American N.S.R.C. The European Broadcasting Union (E.B.U.) has established a working party to consider broadcasting aspects of stereophony. This working party, on which the BBC is represented, is actively engaged in a comparative study of different microphone and transmission techniques, involving, among other things, the exchange of recordings made in the various member countries.

* Details of the membership and constitution of this committee are given in Reference 15.

Broadcasting, as a medium for the dissemination of stereophonic programmes, presents a number of technical and economic problems over and above those encountered in the production of commercial recordings. The principal difficulty is that of providing simultaneous monophonic and stereophonic compatible versions of the same programme without an increase in the number of transmitters, but several other aspects of the subject have to be considered. Although much of the programme material at present broadcast could, as far as technical quality is concerned, be equally well, or in some cases better, produced by pre-recording, certain types of item still require to be transmitted 'live'; with stereophonic programme, various practical problems associated with the disposition and movement of performers in the studio are then aggravated. At the same time, difficulties are encountered in the distribution of the programme over long trunk routes.

The main features of an ideal system of stereophonic broadcast transmission may be summarized as follows; it will no doubt be impossible in practice to fulfil all of these requirements simultaneously.

1. The programme should be capable of transmission on a single radio-frequency channel with no significant increase in bandwidth or decrease in service area.
2. The system should permit reception of stereophonic programmes by means of a monophonic receiver fitted with a relatively cheap and simple adapter; if the listener already has equipment for reproducing stereophonic disks it ought to be possible to utilize this also.
3. Whatever compromise may be made in the transmission system, the stereophonic reproduction should bear comparison with good stereophonic disk recordings reproduced in the listener's home.
4. The programme reproduced on a monophonic receiver should so closely resemble a normal monophonic transmission that the degree of compatibility achieved is, in practice, limited only by the nature of the programme material.
5. Until such time as the microphone disposition and signal processing arrangements involved in originating stereophonic programmes are standardized, the system should operate with any pair of stereophonic signals, however they are derived.

The following section deals briefly with methods of producing stereophonic programme material and reproducing it under domestic conditions, while the remainder of the monograph is devoted to a consideration of the special requirements for broadcast stereophony.

3. Production and Reproduction of Stereophonic Programmes

3.1 *Microphone Systems*

The earliest experiments in two-channel stereophonic reproduction by loudspeakers were carried out with two spaced microphones, having no directional properties except at high frequencies, connected respectively to the

left- and right-hand channels; such stereophonic effects as were achieved were principally due to the difference in the time of arrival of the left- and right-hand signals.

It was later found that improved results could be obtained by making the relative amplitudes of the left- and right-hand signals vary with the direction of sound incidence; it has been found[6] that such a difference in amplitude gives the same effect as a difference in time, such as would arise if the sound arriving at the listener's head had come from some point left or right of centre. A method of operating on the signals from a pair of spaced omnidirectional microphones to produce the required amplitude differences was devised by Blumlein[4] but was later superseded by the present-day practice of employing a pair of coincident directional microphones with their axes inclined to left and right. The latter system is now employed by E.M.I. and has been described in detail by Clark, Dutton, and Vanderlyn.[5] An alternative arrangement of directional microphones[7,8] which produces the same effect, and which has come to be known as the M-S system, will be referred to later in the course of the discussion on compatibility.

A composite system utilizing both time differences and amplitude differences is in use by some commercial recording organizations. The main left- and right-hand signals are obtained from two microphones, each having a cardioid directional characteristic, placed anything between $1\frac{1}{2}$ ft and 10 ft (0·5 m and 3 m) apart; for reasons already indicated, a third, forward-pointing, cardioid microphone is often mounted mid-way between the other two and a portion of its output signal added to each channel.

Another possible arrangement, used by the Philips organization and described by de Boer,[8] consists of a pair of pressure microphones mounted in the 'ears' of a dummy head. This device appears to be identical with that employed by the Bell Laboratories for early work on binaural transmission, i.e. using headphones for reception, and its application to a system employing loudspeaker reproduction is entirely empirical. The microphones employed have no directional properties in their own right, but the obstacle effect of the dummy head produces pronounced directional effects at high frequencies; the system therefore operates in different regimes in different parts of the audio-frequency range.

From the literature it would appear that opinions are divided on the relative merits of the various microphone arrangements described above, but the weight of published evidence appears to be in favour of employing pairs* of closely spaced or coincident directional microphones, while the early practice of using spaced omnidirectional microphones as a stereophonic pair can be regarded as obsolescent. Apart from this conclusion, experience to date suggests that no one arrangement will be equally satisfactory for all purposes, especially when the possibility of special effects in drama is considered, and that it is preferable to use microphones with directional characteristics which can be adjusted as required.

Reference was made in Section 2 to the use, in film production, of the 'pan-pot' technique for utilizing signals from a single microphone to produce a discrete image at any desired point on the sound stage. This artifice is also applicable to two-channel stereophony where it has been found useful in dealing with weak voices or musical instruments requiring special reinforcement, and also in producing moving effects, such as the sound of a trotting horse, from existing monophonic recordings. It has even been proposed that all stereophonic programme material should be synthesized by this means, and an experimental two-channel broadcast transmission on these lines was carried out in France some years ago[9] under the title *Stéréophonie dirigée*. Unfortunately, the stereophonic images produced in this way do not give the impression of breadth and spaciousness obtainable by the twin-microphone arrangement, since it is impossible for any one microphone to separate the reverberant sound from the direct and both are heard coming from the same point; *Stéréophonie dirigée* must therefore be regarded as supplementing rather than replacing the normal stereophonic microphone technique.

3.2 *Disposition of Microphones*

It is only within the last few years that two-channel stereophony with loudspeakers has advanced from the status of a technical 'stunt' to that of a serious artistic medium, and it is therefore not surprising that techniques such as the disposition of microphones in the studio are not yet fully established.

It has often been assumed that all that would be necessary for good stereophonic reproduction would be to place a pair of microphones in a position which would be satisfactory for a listener in the hall. This appears to be an over-simplification, probably based on early experiments with headphones. Although stereophony with loudspeakers restores to some extent the ability to locate the various parts in the performance, a microphone placed in a position which is satisfactory for direct listening often gives results which are too reverberant and sound rather distant. It has further been found that, contrary to various theoretical predictions, the best microphone distance from an orchestra may be less with a stereophonic than with a monophonic system. In all cases, the distribution of the performers and microphones in the studio has to satisfy not only the normal requirements for the balance and perspective of the various instruments but must give an aesthetically satisfactory spatial distribution of the resulting images in the final presentation. This condition is often very difficult to achieve without disturbing the normal orchestral layout; in the case of a light orchestra presenting material of contrasting types, the ideal procedure—hardly practicable except in recorded programmes—is to regroup the players between items.

It would appear in general that whenever possible one main stereophonic microphone pair, corresponding to the single main microphone used in monophonic reproduction, should provide the general sound image with appropriate

* It is necessary here to distinguish between on the one hand a pair of spaced microphones covering the whole working area in the studio and, on the other, two microphones used individually for close-range pickup of particular instruments in the manner described in the next paragraph.

perspective, while subsidiary microphones, whether functioning singly or in pairs, should be used with discretion and only where obviously essential. Although multi-microphone arrangements will still be necessary for dance music, variety music, and drama, the need for special emphasis of soloists will probably be less, other things being equal, than in a monophonic system. As in monophonic reproduction, multi-microphone arrangements may result in incongruous perspectives; in addition, because positional information is transmitted, it is possible for the same artist to appear simultaneously in more than one place on the stage.

Where artificial reverberation is required, this can be introduced by the use of a conventional reverberation room provided with a stereophonic microphone pair. Alternatively, a mechanical reverberation device such as the steel plate described by Kuhl[10] could be adapted for the purpose by providing a second vibration pickup slightly spaced from the first. Other artificial reverberation systems giving only a monophonic output signal cannot, however, produce a natural distribution of reverberant sound across the stage.

3.3 *Pseudo-stereophony*

Attempts have been made by various workers to achieve, with a monophonic signal, something of the spacious effect which characterizes stereophonic reproduction, without, of course, the original positional information; the results thus produced may be described as pseudo-stereophony. Some of the methods employed involve special loudspeaker arrangements and are outside the scope of this monograph. Of particular interest for the present purpose, however, are devices which produce from the monophonic signal two new signals, which, when fed to two loudspeakers set up as for stereophonic reception, give the subjective impression of sound images spread across the entire stage width.

Pseudo-stereophonic effects can be produced by feeding a monophonic signal to the two stereophonic channels through networks so designed that the relative amplitudes and phases of the two outgoing signals, and hence the positions of the resulting sound images, vary with frequency. The directional 'information' thus arbitrarily imposed on the programme, while acceptable on noises with an almost continuous spectrum, can give unwanted effects with music, where the apparent position of the instrument may depend on the particular note being played.[11] However, a device known as the 'Stereophoner', operating on the principle described above, has been devised by the well-known conductor Hermann Scherchen and produced in this country for domestic use. This apparatus is primarily designed for use with symphonic music and the distribution of images is intended to follow as far as possible the normal arrangement of an orchestra, with low-frequency sound emanating from the right, and high frequencies from the left.

Pseudo-stereophonic effects of the kind described have already been found useful in producing special background noises in some stereophonic dramatic productions; a more important potential application could conceivably arise in a stereophonic broadcasting system if there were some part of the programme for which full stereophonic transmission was considered impracticable or unnecessary.

3.4 *Studio Control Equipment*

As a rough approximation, the amount of control apparatus required for a studio used for stereophonic programmes may be arrived at by imagining all channels to be duplicated. In designing the equipment, however, a number of additional requirements must be considered.

In all known stereophonic microphone techniques, it is necessary to provide some kind of electrical adjustment to control the scale of width of the final presentation, i.e. the relationship between the angular displacement of the source and that of the reproduced image. In the case of the three-microphone arrangements the ratio of the contributions made by the centre and side microphones to the outgoing signals represents the variable element, while in two-microphone systems the scale-of-width control is effected by introducing a certain amount of cross-coupling between the left- and right-hand channels.

Where stereophonic effects are based on the difference between the signal amplitudes in the two channels, it has been found[5] that because of subjective factors not yet fully understood, the scale of width is a function of frequency, and in the E.M.I. system an attempt is made to introduce a corresponding correction by means of a phase-compensated equalizer circuit. If different microphone arrangements are to be used simultaneously, corrections of this kind cannot be applied to the final studio output signal but must be provided on individual microphone channels as required.

A number of instrumental refinements are necessary to avoid accidental unbalance between the left- and right-hand channels. In particular, the left- and right-hand faders which control channels, groups, the main studio output, and various other circuits, require to be mechanically linked and to have smaller steps of attenuation than in monophonic equipment to reduce the error resulting from imperfect ganging.

Finally, a number of 'pan-pots', each of which consists of a pair of ganged faders, having a special law of attenuation with angle, and working in opposite directions, are required for use with monophonic sources.

Fig. 1 shows by way of illustration an experimental stereophonic control desk designed in the BBC Research Department on the basis of experience gained in studios during 1958 and 1959; one of the two monitoring loudspeakers can also be seen in the background. In this equipment provision is made for every known system of stereophonic microphone technique, but to allow for future development and for the introduction of special effects, the control desk is built up on a system of interchangeable units. Four channels are provided for stereophonic sources —some of which, in the case of drama, may consist of prerecorded background material—and three for monophonic sources. There is an additional stereophonic channel for artificial reverberation, which is derived from a pair of microphones in a reverberation room. Each

stereophonic channel is provided with a pre-set trimming attenuator to offset any difference in the sensitivities of the left- and right-hand microphones; provision is also made for applying a deliberate left or right bias when required. The 'panning' and scale-of-width controls occupy the bottom row of panels on the control desk; one of the 'pan-pots' is foot-operated, the setting being indicated on a dial.

A number of special effects are provided for use in dramatic productions. The loudspeaker in the reverberation room is normally fed with the sum of the left- and right-hand signals, but by utilizing the left or right signal alone, the reverberation can be confined to sound sources on one or other side of the studio. Again, the output of the reverberation room is normally presented stereophonically so as to cover the full width of the stage, but can instead be made to appear at the left, right, or centre only. There is also provision for producing pseudo-stereophonic effects of the kind referred to in Section 3.3.

3.5 *Studio Acoustics*

It is as yet rather early to determine whether studios used for monophonic purposes will require alteration to their acoustic treatment when employed for stereophony. It is thought that most orchestral studios will be satisfactory without modifications although it may well be that very different microphone positions will be necessary. In studios for light entertainment and drama, multi-microphone techniques are essential to obtain the desired results; experience to date suggests that there may sometimes be difficulty in segregating the various parts of a performance for stereophony, in which case some reduction in reverberation time would be an advantage.

3.6 *Loudspeaker Systems for Domestic Stereophony*

With a few exceptions, equipment at present being produced for domestic reproduction of stereophonic recordings includes a pair of nominally identical loudspeakers intended to be placed between 5 and 10 ft (1·5 and 3 m) apart. As previously indicated, some workers[12] have advocated the use of loudspeakers with a particular form of directivity pattern designed to make the position of the sound images less dependent on the position of the listener. This proposal is, however, difficult to implement in practice and the present trend of design is rather towards reducing directional effects, at least in the horizontal plane, to a minimum.

As an alternative to the straightforward arrangement referred to above, the left- and right-hand loudspeakers are sometimes restricted to the reproduction of frequencies above about 300 c/s, the whole of the low-frequency register being reproduced by a third loudspeaker centrally placed. This simplified form of presentation reduces the cost and bulk of the equipment; the principle involved, however, requires some examination.

It is known that directional impressions are less precise at low frequencies; it has therefore been suggested[13] that in stereophonic transmissions it should be possible to remove all directional information relating to the lower-frequency components without detriment to the final presentation. If this were indeed the case, it would be permissible, not only to simplify the loudspeaker arrangement in the manner just described, but to make the left- and right-hand signals below a certain frequency identical—a procedure which could ease the design requirements in certain of the systems of multiplex transmission discussed in Section 6.3. The suggestion, however, implies that every low-frequency sound is accompanied by such high-frequency components as will suffice in themselves to indicate the proper position of the image. Although this assumption is in most cases correct in respect of the direct sound reaching the microphone, it is not applicable to the reverberant sound, in which the high-frequency components have usually been heavily attenuated in the room and are in any case no longer coherent with the corresponding low-frequency components. Thus, if the directional information at frequencies below 300 c/s is removed, the corresponding components of the reverberant sound, which in good stereophonic reproduction help to give the effect of breadth and spaciousness, are heard coming from a narrow region in the centre of the stage. It is concluded therefore that the principle of removing directional information at low frequencies, while perhaps a legitimate economy measure in the design of reproducing equipment would, if applied at the transmitter, compromise an important advantage of the stereophonic presentation.

4. Fundamental Problems of Compatibility

4.1 *General*

The term 'compatibility' as applied to stereophonic sound transmission is open to varying interpretation according to the aspect of the problem being considered. In general terms, the ideal of compatibility may be defined as the simultaneous transmission of *optimum* stereophonic and *optimum* monophonic versions of the same programme, without the use of additional communication channels and, in the case of the monophonic version, without additional receiving equipment. To achieve this aim a number of conditions, aesthetic and economic, as well as technical, must be fulfilled.

4.2 *Influence of Programme Material*

For a broadcasting organization, the question of compatibility begins with the choice of programme. Most types of programme material at present transmitted monophonically would benefit in some degree from stereophonic presentation, but many of these would not, as they stand, take full advantage of the extra dimension thus introduced. Conversely, programmes designed to exploit all the potentialities of the new medium are likely to lose much of their effect in a monophonic version. Thus, even though all the essential engineering requirements for compatibility are met, the entertainment value of either the monophonic or the stereophonic service or of both can be compromised from the start by the nature of the material transmitted.

4.3 *Derivation of Monophonic Signal*

Assuming 'compatible' programme material, it remains to provide a suitably disposed set of microphones. It sometimes happens that two transmitters, for example, one medium wave and one v.h.f., serve the same area with the same programme, and it might at first sight appear that if each channel were fed from a separate microphone, there would be no difficulty in providing, at one and the same time, a stereophonic transmission and two normal monophonic services. Such an arrangement is, in practice, far from satisfactory, since the two stereophonic signals are derived from microphones which, because of their position or orientation, give prominence to sounds originating respectively on the left or right. In these circumstances, the more the programme exploits the potentialities of stereophony, the more one-sided is the reproduction from either channel taken by itself. There is evidence that some American stations at present using their medium frequency and v.h.f. transmitters to provide a stereophonic service have adopted the expedient of diluting the stereophonic effect, for example, by cross-coupling between the left- and right-hand channels.[14,15] Moreover, it would appear that stereophonic programme material is in some cases selected to avoid poor balance on either channel alone. A system of transmission intended to overcome this difficulty will be discussed in Section 6.2.

Most systems of compatible broadcast transmission are, however, concerned with the problem of utilizing a single transmission channel for simultaneous stereophonic and monophonic transmission; in all the schemes so far proposed, the monophonic signal is derived by the combination—usually by simple summation—of the two* stereophonic signals. It may be noted in passing that stereophonic disks, recorded on the recently standardized system, when played with a monophonic pickup, will likewise reproduce the sum of the left- and right-hand signals.†

In employing a single set of microphones for both monophonic and stereophonic transmission, the possibility of having the optimum microphone *position* for each is ruled out from the start. Moreover, the addition of the left- and right-hand signals gives the effect of a single microphone having directional characteristics which are not always suitable for monophonic transmission. In simple cases, a satisfactory compromise can often be found by suitably modifying the disposition of artistes and microphones in the studio, but there are notable exceptions, particularly in cases where a monophonic transmission would require a multi-microphone arrangement, and these must be taken into account when planning the joint programme.

At this point it seems appropriate to emphasize that a broadcasting organization must be prepared to transmit, in addition to live material from its own studios, recorded material produced by commercial undertakings employing various microphone techniques. Whatever method of compatible transmission is adopted as standard, it should not operate to the disadvantage of any of the known microphone arrangements nor should it place a restriction on subsequent developments in studio technique.

4.4 *Use of Sum and Difference Signals*

Having provided the sum of the left- and right-hand signals for transmission as a monophonic programme, which, with the reservations previously made, might be

* It is assumed that the final stereophonic presentation will require two channels although future refinements might well involve the use of more than two loudspeakers.

† It should be pointed out, however, that with many existing monophonic pickups, having a high mechanical impedance to vertical movement of the stylus, damage to the stereophonic disk will result.

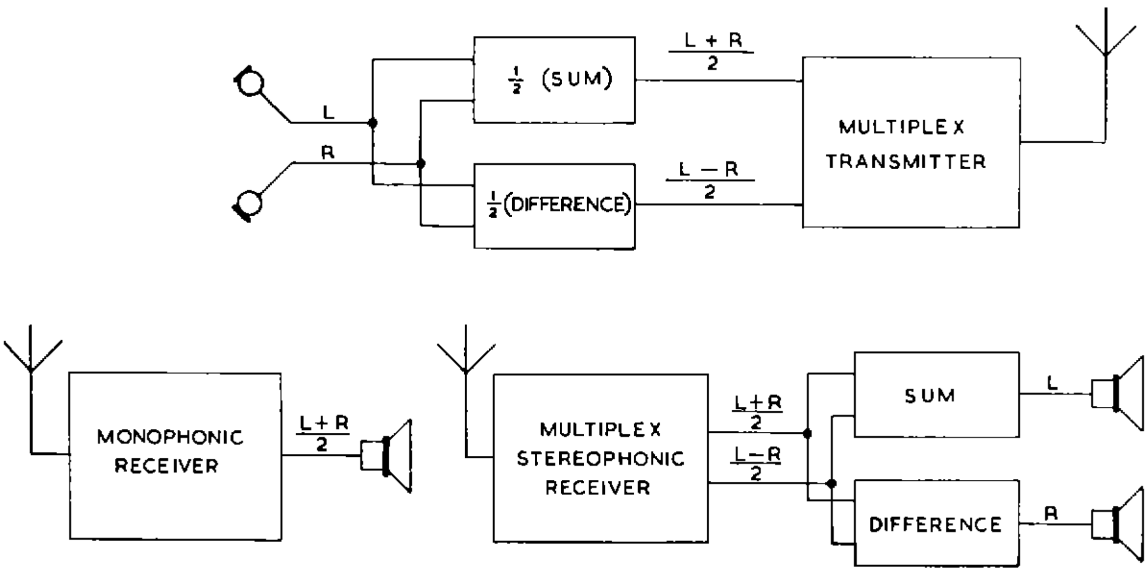

Fig. 2 — Generalized compatible stereophonic system using multiplex transmission

regarded as 'compatible', it is necessary to produce a supplementary signal containing directional information; this can then be transmitted by some form of multiplex system, i.e. as a band of frequencies located outside the normal audio-frequency range or by some other form of modulation which does not interfere with the monophonic reception. Most of the proposed systems of single-channel stereophonic transmission employ, as the supplementary signal, the difference between the original left- and right-hand signals. In a stereophonic receiver, the difference signal is extracted and transposed if necessary to its original audio-frequency band; recombination with the sum signal then yields, by a simple process of addition and subtraction, the original left- and right-hand signals. The whole multiplex process is illustrated in simplified block schematic form in Fig. 2. Various radio-frequency transmission systems operating on this principle are discussed in Section 6.

In systems of stereophony employing a pair of identical, coincident, directional microphones inclined to left and to right, the sum signal could equally well be obtained directly from one microphone pointing forward and the corresponding difference signal from a second microphone, having a figure-of-eight polar pattern, turned at right angles to the first. (There is, of course, no saving in equipment, since a sum-and-difference operation has still to be carried out at the studio to obtain left- and right-hand signals for monitoring purposes.) This alternative method of deriving sum and difference signals, which has been known since the earliest days of the art,[4] has recently been revived by Lauridsen[7,8] and has come to be known as the M-S or Mid-Side system. It is important to note that the use of forward-facing and figure-of-eight microphones in this way rather than a pair of symmetrical microphones in conjunction with a sum-and-difference network involves no new principle. With modern techniques the two arrangements can be regarded as equivalent and, in particular, neither can be said to give a greater degree of compatibility than the other.

5. Problems of Programme Distribution by Line

5.1 *General*

In stereophonic reproduction the position of the sound image depends upon the relative amplitudes and phases of the sounds received by the observer from his two loudspeakers. Accurate localization of the image is therefore conditional on accurate matching of the characteristics of the two complete transmission chains from microphone to loudspeaker. In so far as this requirement is not fully met—for example, through the use of loudspeakers of dissimilar type—the position of a given musical instrument or artist may not be sharply defined and the image may be displaced to left or right. It is true that under these conditions, the general effect of spaciousness which characterizes stereophonic reproduction is retained to a surprising extent, while simple effects which depend on the sound coming from either the extreme left or extreme right are not seriously impaired.* However, such a 'quasi-stereophonic' effect, while generally regarded as preferable to monophonic reproduction, is substantially inferior to the result obtained with accurate matching of the two transmission chains and would not bear comparison with good domestic stereophonic gramophone reproduction.

In broadcasting, the most difficult links in the chain to match are the land lines from the studio to the transmitter. These lines are usually not less than 10 miles (16 km) long and within the United Kingdom may be as long as 800 miles (1 300 km).

5.2 *Effect of Varying Loss in Lines*

The stability of the lines and equipment in the present programme distribution chain should be sufficient to allow the overall mid-band loss established in the daily line-up routine to be maintained within ± 1 dB throughout the programme time. Any greater deviations than this are generally attributable to accidental disturbance of gain adjustments or to regrouping of circuits to meet the exigencies of the service and are dealt with by station staff on an *ad hoc* basis.

If two such lines are employed to transmit the left- and right-hand signals of a stereophonic programme, the two mid-band losses may differ by 2 dB. This difference will produce a sideways displacement of all sound images in the central part of the stage to the extent of about 7 per cent of the stage width.

5.3 *Effect of Differing Frequency Characteristics*

The normal tolerances on frequency characteristics of programme distribution circuits allow for an 800 miles (1 300 km) route a total spread of 4·5 dB over the range 50 c/s to 7 kc/s. Even if the deviations were distributed equally above and below the mid-band value, two such lines adjusted to have equal mid-band loss could differ in attenuation by 4·5 dB in particular parts of the band;† this disparity alone could produce a total image displacement of about 16 per cent of the stage width.

5.4 *Effect of Differing Group-delay Times*

The group-delay times on programme lines within Great Britain may be as much as 20 ms in the middle of the audio-frequency range, rising by perhaps 30 per cent at the upper end of the frequency band and by a considerably greater amount at very low frequencies. Unless special precautions are taken, two such lines, running over the same nominal route, may differ in time delay by several milliseconds.

The primary effect of this difference in delay is to displace the stereophonic images to left or right. It must be remembered, however, that the stereophonic signals reach the broadcast listener through channels whose gain is not fixed. The gain controls associated with the two loud-

* These remarks apply to a large extent to the reception, outside the London area, of the fortnightly experimental transmissions started by the BBC in October 1958.

† The fact that the frequency characteristics of nominally similar loudspeakers may exhibit local deviations of perhaps ± 5 dB does not diminish the significance of the broad trends in response associated with the line characteristics.

speakers have in practice to be adjusted by ear, the listener unconsciously compensating* as best he can for the difference in delay by decreasing the gain of that channel of which the signal is in advance, or vice versa. Unfortunately, the rate of exchange between attenuation and time difference is by no means a constant, so that after the preliminary adjustment has been carried out, the following phenomena are observed—the figures given refer to the group delays at mid-band.

For differences in delay up to about 0·25 ms, the effect on the stereophonic presentation is a slight displacement of some of the images, which would probably not be noticed except by direct comparison with the original condition.

As the relative delay approaches about 0·5 ms, a change of regime begins. The images in the centre of the stage are less sharply located and often appear more distant; the correct relative gain and polarity of the two channels become less clearly defined and may differ with different types of programme.

At about 2 ms relative delay, the change of regime is virtually complete; the delay can then be increased to as much as 10 ms without further deterioration in the situation, though with particular values slightly better or worse results may be obtained.

It would appear from the above that the mid-band group delays of a pair of stereophonic transmission channels should not differ by more than about 0·25 ms. To establish and maintain this condition on lines of the present type, while technically feasible, would be a laborious and costly business, the more so because in practice the various links of the chain are liable at times to be rearranged to suit the exigencies of the service.

If, however, the use of a compatible radio-frequency transmission system involving sum and difference signals is considered, the requirements for matching of line delays are seen to be even more stringent. If the summation of the left- and right-hand signals is carried out at the input to the transmitter, cancellation effects arising from differences in the phase delay of the incoming lines can produce serious irregularities in frequency characteristic, with nulls at intervals. With 0·25 ms delay difference, for example, the first null will occur at 2 kc/s, while if a null is not to occur below 8 kc/s, the maximum allowable delay difference at high frequencies would be 0·06 ms.

If in an attempt to circumvent this difficulty the sum signal is transmitted on one line and the difference signal on the other, the interference effects are absent from the monophonic programme but appear instead in the received stereophonic programme, being at their worst with sound images for which either the original left- or right-hand signal is zero. Further, at any frequency for which a complete phase inversion of the difference signal takes place, the stereophonic presentation is reversed left to right. It should be noted that in any part of a stereophonic transmission system in which the sum and difference signals are separated, quite small disparities between the amplitude or phase characteristics of the two channels concerned will give the effect of cross-talk between the left- and right-hand signals. In a broadcasting system, it is suggested that the cross-talk attenuation should in general be at least 20 dB, to which end the sum and difference channels should not differ by more than 1·8 dB in attenuation or 11° in phase. Since both amplitude and phase difference will in general occur simultaneously, smaller tolerances will in practice be required.

All that has been said on the subject of the permanent line distribution system applies in some measure to the lines employed for outside broadcasts. Although the distances involved are in general less, the programme is fed into the main distribution network through temporary land lines whose characteristics cannot be so conveniently controlled as those of the permanent routes.

5.5 *Alternative Methods of Distribution*

In view of the foregoing it would appear that the present system of programme distribution by lines is not suitable for the distribution of stereophonic programmes on a compatible basis. For a nation-wide service of this kind, including live programmes, alternative methods of distribution, such as carrier systems or radio links, may have to be considered. In the meantime, there is a good case for making use wherever possible of live or recorded programmes† originating at distribution centres located not too far from transmitters, to avoid sending stereophonic programmes over long trunk routes.

6. Systems of Transmission

6.1 *General Considerations*

To present a summary of various methods for transmitting stereophonic programme on available radio channels is rather difficult at present because of the rapid development. In the United States the National Stereophonic Radio Committee mentioned in Section 2 had by March 1959 received for consideration proposals for seventeen systems covering applications to television, a.m. and f.m. broadcasting. The information available on most of these and on other systems being developed in Europe is rather sketchy, and it must suffice here to single out a few of the basic ideas which underlie the most promising systems. A useful summary of various systems, principally of American origin, has recently been published.[16]

In the United States, at least among the broadcasting organizations, it is considered essential that an acceptable system of stereophony should be evolved for a.m. transmission in the m.f. broadcast band. This may be additional to higher-quality systems for v.h.f. sound transmissions, but there are three factors which may cause m.f. developments to proceed further in the U.S. than in Europe, namely (*a*) the present use of the m.f. band as the only medium of sound broadcasting over considerable areas, (*b*) the relatively less congested state of the m.f. band, and (*c*) the present practice of v.h.f. stations of augmenting

* Because of this compensation, the results which follow are not to be confused with the effect of simple displacement of the observer to left or right of the centre line.

† To make full use of the technical possibilities of recording, for example by reproducing recordings of the same programme in different studio centres, would, in the U.K., involve revision of some existing contractual arrangements.

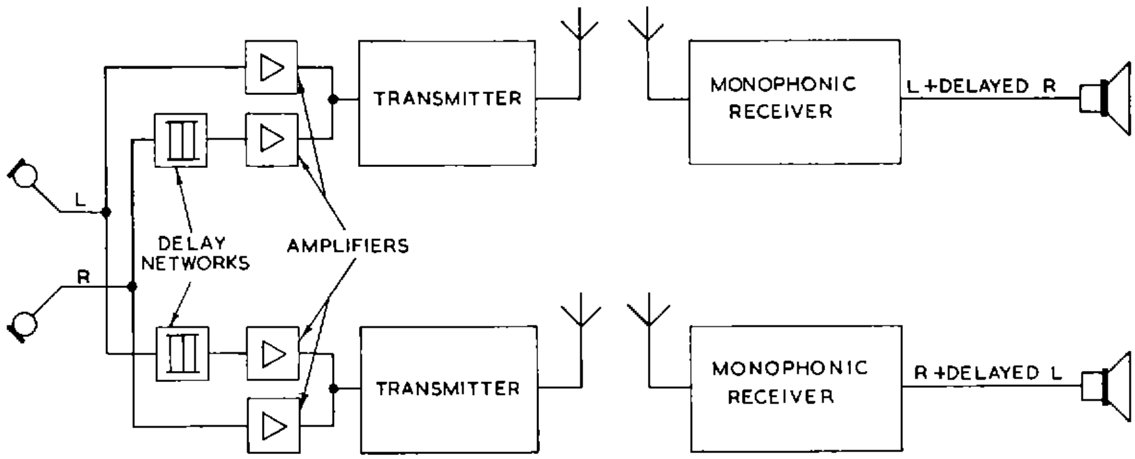

Fig. 3 — Bell system using individual stereophonic channels for monophonic reception

their income by providing subsidiary multiplex services, thus preventing the fullest possible use of multiplex techniques for stereophony in the v.h.f. sound broadcasting band.

6.2 *Systems Requiring Two Radio Frequency Channels*

The most direct method of stereophonic broadcasting is to allocate two separate broadcasting channels to the left-hand (L) and right-hand (R) signals. This in fact has been the method used by the BBC in the initial broadcast experiments in the United Kingdom; the L channel has been radiated by all transmitters in the third sound-programme network (reception on either m.f. or v.h.f. being possible) and the R channel has been radiated by the sound channels of the BBC television network. Although the service areas of each individual channel, and hence of a stereophonic broadcast, can be the same as for monophonic broadcasting, the ordinary listener, for reasons already indicated in Section 4.3, will not in general receive an acceptable programme from any one transmitter.

A proposal to overcome the latter difficulty has been made by Becker of the Bell Telephone Laboratories,[17] and is illustrated in Fig. 3. It will be seen that the signal from the incoming left-hand channel is fed into the outgoing right-hand channel through a network giving a slight time delay—something between 5 ms and 30 ms is suggested—and vice versa. Each outgoing channel now contains information from both sides of the system, and if equal proportions of the two are combined, the monophonic programme signal is, as far as balance is concerned, the sum of the left and right signals. When both channels are reproduced simultaneously for stereophonic reception, the positions of the sound images are determined, other things being equal, by the sound which first reaches the listener's ears, a phenomenon known as the Haas effect or 'precedence' effect. In these circumstances the presence of the delayed signals does not interfere with the stereophonic presentation. A slight modification of this system, in which the cross-mixed signals are delayed by only 3 ms but attenuated by as much as 6 dB, has been reported, but any attenuation naturally reduces the effectiveness of the device in avoiding unbalance. In more recent experiments a delay of 10 ms and an attenuation of only 1·5 dB have been employed but the optimum values of delay and attenuation depend to some extent on the nature of the programme material.[18]

It is an inescapable feature of the system described that whenever a sound source produces signals of comparable amplitude in both the incoming channels, each outgoing channel will be transmitting two versions of the same sound slightly separated in time; when this happens there is inevitably a certain deterioration in quality. With delays of a few milliseconds, there is a colouration of the sound, usually described by observers as a 'honk' or 'twang', while with values increasing above 20 ms, the added sound begins to resemble a single echo; the least objectionable results are obtained with intermediate values of time delay. The success of the system is therefore conditional on whether the best compromise between quality and balance is acceptable.

6.3 *Multiplex Systems Providing Two A.F. Channels on a Single R.F. Channel*

In the discussion of compatibility in Section 4 it was pointed out that a $\frac{1}{2}(L+R)$ signal (or some other acceptable combination of the L and R signals) must be transmitted by the standard method of modulation in order to give a satisfactory monophonic service. Any compatible multiplex system which provides $\frac{1}{2}(L+R)$ for this service can therefore be regarded as dealing basically with sum and difference signals as indicated in Fig. 2. There have from time to time been suggestions that satisfactory results are obtainable when the difference signal has not the full a.f. bandwidth of the sum signal; for example an upper-frequency limit of only 2 kc/s has been suggested in one instance,[19] while in another case a lower frequency limit as high as 500 c/s has been suggested.[20] Sounds in the parts of the frequency range in which the two channels are effectively commoned by curtailment of the difference signal are reproduced in equal strength by the two loudspeakers, and the resulting images are then confined to the central region of the stage. The disadvantages in the case of

TABLE I

Systems for A.M. Transmissions

Systems and proposers	*Modulation with L signal present but no R signal*	*Modulation with $\frac{1}{2}(L-R)$ signal but no $\frac{1}{2}(L+R)$ signal*	*Frequencies for which $\frac{1}{2}(L-R)$ channel is curtailed*
(*a*) Philco Corporation, Columbia Broadcasting System	a.m. and ph.m. in step	pure ph.m. ($\pm\pi/4$ radian peak)	<300 c/s
(*b*) R.C.A. (1958), Kahn Laboratories	single sideband (a.m. and ph.m. in quadrature)	pure ph.m. ($\pm\pi/4$ radian peak)	not known
(*c*) Westinghouse, General Electric	a.m. and f.m. in step	pure f.m. (±3 or 4 kc/s peak)	<300 c/s $>3{,}000$ c/s*

* This limit applies to Westinghouse.

the lower audio frequencies were mentioned in Section 3.6, while the loss of stereophonic effect on many high-pitched sounds is certainly very noticeable with wide-range equipment when the upper frequency limit of the difference signal is of the order of 5 kc/s. Attention is therefore being paid, both in Europe and in the U.S.A., to multiplex systems providing two audio-frequency channels with no degradation in the frequency range relative to normal transmissions.

The systems to be considered for sound broadcasting divide naturally into those proposed for m.f. broadcasting, being suitable only for a.m. transmissions, and those proposed for v.h.f./f.m. broadcasting. Developments in systems for television sound channels have not proceeded very far, but suitable systems would probably be simple adaptations of those for either a.m. or f.m. transmissions, whichever is appropriate; they would not be likely to make use of the vision signal apart from the possibility of employing the line time base as a reference oscillator in systems requiring synchronized detection.

6.3.1 *Multiplex Systems Suitable for A.M. Transmissions*

Considerable interest exists in the United States in methods of compatible stereophonic broadcasting over standard a.m. channels in the m.f. band. As far as is known at present, all the multiplex systems proposed conform in principle to the straightforward scheme of Fig. 2 and provide $\frac{1}{2}(L+R)$ as amplitude modulation. There are, however, a number of different proposals regarding the effective method of conveying the difference signal, though they are all based on the use of frequency or phase modulation of the main carrier as a function of the difference signal. The basic proposals are summarized in Table 1 by giving the type of modulation transmitted for two special conditions. In all cases, standard double-sideband a.m. is transmitted when the difference signal is zero.

In System (*a*), if a signal exists in one channel only, the end point of the resultant r.f. vector moves on a slewed path relative to the carrier vector, as shown in Fig. 4. It is possible to generate this signal by combining the outputs of two ordinary a.m. transmitters in which the carrier drives are identical in frequency but are in quadrature phase; one transmitter is modulated by the L signal and the other one by the R signal. With only one of the L or R signals present the resultant signal still contains equal *power* in the two sidebands, but the sidebands have a quadrature phase relationship. In general, the system requires the end point of the r.f. vector in the diagram to be displaced from the no-signal position by a distance proportional to the L signal in a direction parallel to one 45° axis and by a distance proportional to the R signal parallel to the other 45° axis. The amplitude gives the $\frac{1}{2}(L+R)$ signal but, except for the special case of $L=R$, with some non-linear distortion which increases with depth of modulation. Thus the permissible depth of modulation must be a compromise depending on the distortion acceptable in monophonic receivers with envelope detectors.

The latter difficulty exists also in the case of System (*b*), which has the property that one sideband carries the L signal and the other sideband the R signal, but it is believed that R.C.A. has made proposals for reducing the distortion in an envelope detector by a form of pre-correction. In the Westinghouse system it is proposed to frequency modulate the drive according to $\frac{1}{2}(L-R)$ and to arrange by direct modulation that the amplitude of the transmitter output represents $\frac{1}{2}(L+R)$. An extra amplitude modulation component, which is a function of the frequency deviation, is applied to compensate for the effect of the average receiver i.f. response, thus reducing distortion of the received a.m. signal in the presence of f.m.

A stereophonic receiver for System (*a*) can use either a single phase-sensitive detector and an amplitude detector, giving $\frac{1}{2}(L-R)$ and $\frac{1}{2}(L+R)$ respectively, or else two phase-sensitive detectors which, with the appropriate reference phases, can give L and R directly. Non-linear distortion is avoided in the latter case; it can also be avoided in a monophonic receiver if phase-sensitive detection is used. A receiver for System (*b*) may derive the L and R signals in one step by using two filters (each rejecting one sideband) followed by two envelope detectors. In System

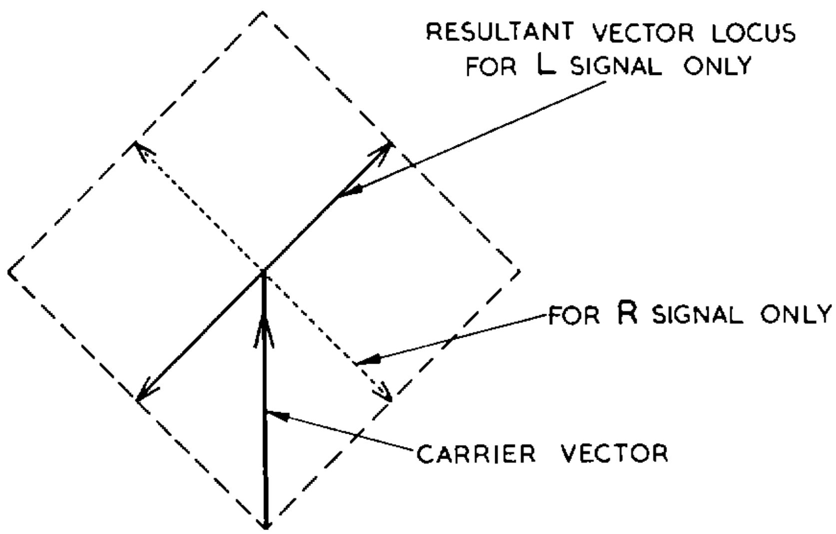

Fig. 4 — The Philco system for a.m. transmissions

(*c*) the difference signal may be obtained directly by a frequency discriminator, and combined according to the scheme of Fig. 2 with the sum signal provided by an envelope detector. Stereophonic reception, though not of high quality, is also possible with System (*c*) from two ordinary receivers slightly detuned in opposite directions.

Since 1958, the R.C.A. has moved away from System (*b*) and has made a new proposal similar to System (*c*). However, high-frequency pre-emphasis of the $\frac{1}{2}$(L—R) signal is used before modulation, with the result that at low frequencies (below about 1·5 kc/s) it corresponds to System (*c*), while at higher frequencies it operates approximately as System (*a*). The receiver requirements are the same as for System (*c*) provided that appropriate de-emphasis is applied to the f.m. discriminator output.

The Columbia Broadcasting System has put forward a proposal corresponding to System (*a*) in the form already described. However, towards the end of 1959, the Philco Corporation proposed a modification to their original system. The $\frac{1}{2}$(L—R) signal is still carried by suppressed-carrier amplitude modulation of a component in quadrature with the main carrier, but the carrier envelope rather than the in-phase component is now arranged to follow accurately the $\frac{1}{2}$(L+R) signal. This eliminates the distortion in monophonic reception, the preferred method of stereophonic reception now being the one employing an envelope detector for the $\frac{1}{2}$(L+R) signal.

The performance of all of these systems in regard to signal-to-noise ratio is not expected to be more than about 3 dB worse than that of normal a.m. transmissions for either monophonic or stereophonic reception. Fuller details of the systems would be needed to be able to give an accurate indication of their performance.

6.3.2 *Multiplex Systems Suitable for F.M. Transmissions*

At present it is common practice in U.S.A. for f.m. stations to use a frequency-modulated subcarrier to convey an extra programme or service, such as background music. This is a private subscriber service to stores, restaurants, and similar establishments, thus providing useful revenue for the station. In some cases two separate services of this kind besides the main broadcast programme are radiated

TABLE 2

Systems for F.M. Transmissions using an F.M. Subcarrier

Alternative systems	(*a*)	(*b*)	(*c*)
Subcarrier frequency	45 kc/s	50 kc/s	50 kc/s
Peak deviation of subcarrier by the $\frac{1}{2}$(L—R) signal	±15 kc/s	±25 kc/s	±25 kc/s
Peak deviation of main carrier (% of ±75 kc/s)			
(1) by the $\frac{1}{2}$(L+R) signal	50%	50%	70%
(2) by the subcarrier	50%	50%	30%
Theoretical signal-to-noise ratios relative to a standard ±75 kc/s f.m. broadcast			
(1) $\frac{1}{2}$(L+R) channel	—6 dB	—6 dB	—3 dB
(2) $\frac{1}{2}$(L—R) channel	—18·5 dB	—15 dB	—19·5 dB
(3) L or R channels	—16 dB	—12·5 dB	—16·5 dB

from one station, using subcarrier frequencies of 41 kc/s and 67 kc/s. The 'Halstead Stereoplex' system has been proposed which uses the main channel for the L signal and one of the subcarriers (41 kc/s) for the R signal, the second subcarrier being used for the subscriber service. This system encounters the same difficulty in regard to compatibility as one using two radio channels, and a process of cross-mixing is used in an attempt to overcome this objection. While the system is attractive to stations who are, not unnaturally, reluctant to give up their subscriber service, it makes rather serious compromises and must inevitably accept an appreciable reduction in the bandwidth and/or signal-to-noise ratio of the subcarrier channel. It is not therefore expected to be of direct interest in Europe at the present time. The omission of the subscriber service would allow the remaining subcarrier channel performance to be improved, but the difference in performance between the L and R channels and the difficulty of compatibility would remain.

The most promising systems are confined to those in which a single subcarrier modulated by a $\frac{1}{2}$(L−R) signal, is added to a $\frac{1}{2}$(L+R) signal at the modulation input to the main f.m. transmitter. They may be subdivided into those in which frequency modulation of the subcarrier is used, and those which effectively use amplitude modulation of the subcarrier. The latter may include switching or 'time-division multiplex' systems, since bandwidth restriction usually makes it easier to regard them as subcarrier systems.

(*i*) *F.M. Subcarrier Systems*

Crosby[21] in the U.S.A. has proposed a system employing a subcarrier which is frequency modulated by the $\frac{1}{2}$(L−R) signal and added to the audio-frequency signal corresponding to $\frac{1}{2}$(L+R). This composite signal is then supplied to the main f.m. transmitter, the levels of the two components being controlled to confine the resultant peak deviation to ±75 kc/s.

The standards have changed slightly[22] since the original proposal. Table 2 gives the details and the theoretical performance in regard to signal-to-noise ratio for the original 1953 proposal under (*a*) and for the 1958 proposal under (*b*); it has also been suggested that the later Crosby proposal could be modified as given under (*c*). Some experiments by the Nederlandse Radio-Unie with a f.m. subcarrier system using slightly different standards from those given in Table 2 have recently been described.[23,24]

(*ii*) *A.M. Subcarrier and Time-division Multiplex Systems*

In a two-channel time-division multiplex system a single communication channel is switched alternately between two signals, L and R. If this is applied to a f.m. system, the instantaneous frequency transmitted would, under idealized conditions, take the form shown in Fig. 5.

In this system it is necessary for the switching frequency, taken as the frequency of a complete switching cycle, to be at least twice the highest frequency in the a.f. band in order to avoid the production of audible beat tones between the switching and signal frequencies. A switching frequency of about 40 kc/s is therefore reasonable. However, it is then not practicable to transmit harmonics of the switching frequency because frequencies of some 80 kc/s or more, applied to the f.m. transmitter in addition to audio-frequency modulation, will give rise to considerable sideband energy well outside the normal frequency channel, and this is to be avoided if possible.

In effect, therefore, the system is equivalent to one using a subcarrier of 40 kc/s, which is amplitude modulated in proportion to the difference signal $\frac{1}{2}$(L−R), together with an audio-frequency signal $\frac{1}{2}$(L+R) representing the mean of the two channels, the combined signal being applied to a f.m. transmitter. It should be added that the amplitude modulation of the subcarrier is of the carrier-suppressed type and it is therefore necessary to transmit a reference tone either at 20 kc/s or by a residual 40 kc/s subcarrier; the deviation for this can, however, be quite small.

Table 3 gives, under (*a*), the theoretical signal-to-noise ratio for a suitable system. The switching has been assumed to take place between the L and R signals after pre-emphasis with 50 μs time constant, the level of each of these signals being such that it would fully modulate a f.m. transmitter if it were switched permanently through to the transmitter. The modulated subcarrier has been taken to be quasi-sinusoidal, having a peak amplitude equal to that of the quasi-square wave in Fig. 5.* In practice, the method of

* A possible alternative is to allow the fundamental component of the square-wave to represent the subcarrier amplitude, but a peak deviation slightly greater than ±75 kc/s would result. If, however, this is permissible an improvement of 2 dB in the signal-to-noise ratio of the stereophonic channels would be obtained.

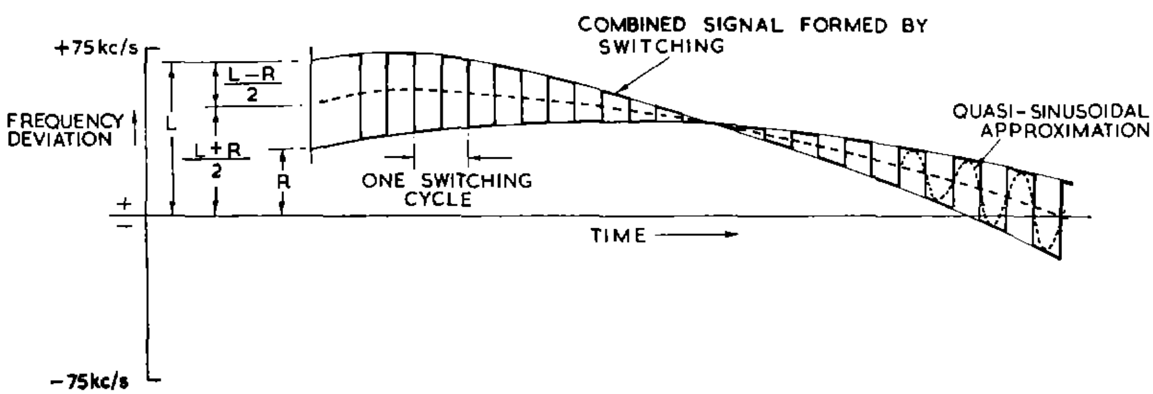

Fig. 5 — Idealized time-division multiplex system

generation of the signal could equally well be one in which the difference signal $\frac{1}{2}$(L−R) is applied to a balanced modulator, using a 40 kc/s carrier, the output being added in a suitable proportion to the sum signal, $\frac{1}{2}$(L+R). Because the noise spectrum of the difference channel differs from that of the sum channel, Table 3 gives the results of calculations in which the noise was weighted according to the C.C.I.R. (1934) aural-weighting recommendation.

TABLE 3

Systems for F.M. Transmissions using an A.M. Subcarrier (subcarrier frequency 40 kc/s)

Theoretical aurally weighted signal-to-noise ratios relative to a standard $\pm$75 *kc/s f.m. broadcast*	
(*a*) using d.s.b. a.m. (carrier-suppressed)	
(1) $\frac{1}{2}$(L+R) channel	−3 dB
(2) $\frac{1}{2}$(L−R) channel	−28·5 dB
(3) L or R channels	−25·5 dB
(*b*) using s.s.b. a.m. (lower sideband)	
(1) $\frac{1}{2}$(L+R) channel	−3 dB
(2) $\frac{1}{2}$(L−R) channel	−25 dB
(3) L or R channels	−22 dB

NOTES: (1) The peak deviation of the main carrier by the $\frac{1}{2}$(L+R) signal is 71 per cent of $\pm$75 kc/s (for L and R uncorrelated), the subcarrier signal being adjusted for $\pm$75 kc/s total peak deviation for system (*a*).

(2) If a frequency of 32 kc/s in place of 40 kc/s can be used for the subcarrier, the signal-to-noise ratio for the difference and stereophonic channels can be improved by about 2 dB.

(3) The figures do not allow for the small proportion of main carrier deviation allocated to the subcarrier reference signal; the degradation of all figures in Table 3 to allow for this is expected to be less than 1 dB.

Systems which use a double-sideband a.m. subcarrier have been proposed by the Zenith Radio Corporation in the U.S.A. and by Grundig in Germany.

A modification of the system just described is to limit the bandwidth of the modulating signal by means of a low-pass filter with a cut-off frequency close to the subcarrier frequency. The upper sideband of the subcarrier signal is then removed, leaving single sideband modulation. By a suitable increase in the level of the subcarrier signal it can be made equivalent in modulation depth to the double sideband a.m. subcarrier signal previously considered. The theoretical signal-to-noise ratio on the stereophonic channels is then improved as indicated in Table 3 under (*b*) assuming that noise in the subcarrier band above the subcarrier frequency is removed in the receiver by a filter. Systems of this type have been put forward by Siemens[25] in Germany and by the Philco Corporation[20] in the U.S.A.; subcarrier frequencies near to 30 kc/s are used in each case.

Provided that the subcarrier signal bandwidths are limited to the minimum required to convey the signals according to the systems described in Table 3, the method of generation could equally well be by the use of short-pulse sampling of the L and R signals alternately instead of by square-wave switching as assumed. However, a proposal by Mullard in the U.K. arranges that L and R signals are unipolar before sampling (i.e. they are accompanied by a d.c. bias so that they are both always positive) and employs half-sine-wave sampling pulses. The subcarrier signal then includes a frequency component of twice the switching frequency, and this is retained in the transmitted signal. In addition, a small synchronizing signal is transmitted at the switching frequency itself, in phase-quadrature with the basic reference carrier. This composite signal can simplify receiver design, but the theoretical signal-to-noise ratio cannot be quite as good as the systems of Table 3 if the total deviation of the main carrier remains within the limit of $\pm$75 kc/s.

The use of an amplitude modulated subcarrier, without using suppressed-carrier techniques, has also been proposed for conveying the $\frac{1}{2}$(L−R) signal with the object of simplifying receiving equipment as far as possible. The complete modulating waveform comprises a $\frac{1}{2}$(L+R) signal combined with this subcarrier signal. A circuit tuned to the subcarrier frequency followed by a simple envelope detector may then be used after the main discriminator to derive the $\frac{1}{2}$(L−R) signal; alternatively the positive and negative envelopes of the composite waveform may be detected to yield the L and R signals directly. Proposals for transmitting systems of this type have been made by the General Electric Company in the U.S.A., by Philips[26] in Holland, by Radiodiffusion-Télévision Française in France, and by Loewe in Germany. Also, some experimental investigations have been reported by the Nederlandse Radio-Unie,[24] which has shown that, as might be expected, the performance of this simpler type of system is not as good in several respects as that of the more sophisticated systems.

(iii) Limitations of Theory

We have so far considered the basic multiplex systems for f.m. transmitters and have given certain theoretical figures without reference to the practical side of receiving equipment. At this point it is convenient to discuss the instrumental difficulties which may affect the performance of receivers and hence the quality of both the monophonic service and the stereophonic service. In general it should be noted that the theoretical figures given for the relative signal-to-noise ratios are applicable only to receiver background noise or to continuous forms of wide-band interference. In practice, the limitation of service is rarely receiver noise but various types of external interference, particularly impulsive interference. The figures given can only form a very approximate guide for the latter type of interference.

In calculating the relative signal-to-noise ratios, using the performance on a standard f.m. transmission as the reference, it was assumed throughout that there is perfect suppression of amplitude modulation in the receiver; in

the case of stereophonic receivers the amplitude limiting circuits before the first detector would require to be efficient at frequencies up to about 75 kc/s, not merely at audio frequencies, in order to give the theoretical performance. Moreover, when quite a large proportion of the available ± 75 kc/s peak deviation is taken up in transmitting a subcarrier, the background noise level in ordinary receivers may rise slightly whereas in the idealized case considered it does not; this rise has been found experimentally to occur in many receivers both for continuous noise and in the presence of impulsive interference. As a result, the degradation in the main channel signal-to-noise ratio on a stereophonic transmission may not simply correspond to the reduced deviation available to the main channel, as indicated in the tables, but may be more serious. This effect is more important for a f.m. subcarrier system in which 50 per cent f.m. at a supersonic frequency is present continuously on the main carrier; preliminary experiments suggest a typical figure of 10 dB rather than 6 dB for the deterioration of the ratio of signal to background noise or impulsive interference. The disparity between the practical and theoretical figures would be a bit less for a system with 30 per cent f.m. allocated for the subcarrier. The effect would be least serious—possibly quite negligible—for the a.m. subcarrier systems of Table 3, since the subcarrier amplitude is automatically reduced to a small value during pauses or quiet parts of the programme being transmitted.

Again, it has been assumed that the L and R signals have equal power but are in a random relationship and, as a consequence, that the levels of $\frac{1}{2}(L+R)$ and $\frac{1}{2}(L-R)$ are equal and are each 3 dB below that of L and R by itself. If, as was originally suggested by Crosby,[21] the L and R signals were well correlated, the difference signal would be much smaller than the sum signal, and a significant improvement in performance could be made. Thus, in the case of the f.m. subcarrier systems, the difference signal could be transmitted at a higher level than has been assumed and correspondingly attenuated in the receiver, resulting in a better signal-to-noise performance on the stereophonic channels. In the a.m. subcarrier systems on the other hand, the transmitted $\frac{1}{2}(L+R)$ signal would be at a higher level than has been assumed, thus giving a monophonic performance rather better than 3 dB below the performance on a standard f.m. broadcast. In fact, analysis of a wide range of recorded material shows that the separation in crest value between the sum and difference signals commonly varies between zero and 10 dB, with an average of some 4 dB; this conclusion is supported by a more recent statement by Crosby.[27] Another factor is that the correlation may be substantially reduced when the programme is distributed, e.g. by line, owing to unequal phase-shifts at high audio frequencies. Any gain in performance over that calculated for uncorrelated L and R signals would not therefore be expected to be very great.

The importance of the two factors discussed in the previous paragraphs, one suggesting a lower and the other a higher signal-to-noise ratio than for the idealized case, requires further investigation.

In the systems compared in the tables, the total peak deviation of the main carrier is limited to ± 75 kc/s. Since modulation frequencies above 15 kc/s are present there will, however, be appreciable energy outside a ± 90 kc/s band of frequencies. It will be noticed that the highest subcarrier frequency component is about 75 kc/s for the f.m. subcarrier system (as proposed in 1958) but only 55 kc/s or 40 kc/s for the two a.m. systems. It is therefore possible that, for the standards given, the energy outside a ± 90 kc/s frequency band is more important in the case of the f.m. subcarrier system than the a.m. systems. Some revision of standards or the incorporation of special filters in the transmitter may therefore be necessary before the comparison of systems on the basis of, say, equal adjacent-channel interfering power can be made.

The presence of a subcarrier modulation component can also affect the severity of co-channel interference. Thus, not only may co-channel interference become more serious with the use of multiplex systems because of the reduced deviation available to the $\frac{1}{2}(L+R)$ signal, but the possibility of reducing interference by an offset of 20 kc/s or more between the main carrier frequencies of two stations may no longer exist with many of the multiplex systems, owing to the increased sideband energy associated with a subcarrier.

A compromise can clearly be made in the case of the f.m. subcarrier systems between monophonic and stereophonic services (as exemplified by systems (*b*) and (*c*) in Table 2) by varying the proportion of the total deviation of the main carrier allocated respectively to the main and subcarrier signals. It is worth noting that a similar compromise is also available in the a.m. subcarrier system. Although the time-division multiplex approach leads naturally to a particular level of modulation for both the sum and the subcarrier signals, the relative levels of these signals can be adjusted prior to modulation of the main transmitter. This is possible either by differential attentuation in a filter or, in the case of the balanced modulator method, by direct control of the sum and difference levels.

(*iv*) *Stereophonic Receivers for F.M. Transmissions using Multiplex Systems*

Turning now to the case of special receivers or adapters for f.m. stereophonic broadcasts it seems fair to say, first of all, that it is too early to indicate until more experimental work has been done how the complexity and performance of these will influence the choice between various possible systems. The present position is that sufficient experience has been gained in the U.S.A. with f.m. subcarrier transmissions of a subsidiary programme to show that the receiving equipment need not be very elaborate. The circuit is conventional up to the main discriminator, and generally employs a subcarrier signal amplifier with a pulse-counter type of discriminator to give the subcarrier signal. Experiments in Holland[23,24] have confirmed that the cross-talk between the main and subcarrier channels can be kept quite small. For application to stereophony, using the sum and difference technique, the a.f. components from the main and subcarrier discriminators must finally be combined in an additive and subtractive manner in order to derive the L and R signals.

The receiving equipment for the a.m. subcarrier sys-

tems is similar apart from the subcarrier demodulator. Since the most promising systems employ a suppressed-carrier type of modulation the receiver must have some form of oscillator at the subcarrier frequency synchronized by the relatively small reference signal. Whether this leads to greater complexity and cost in a complete receiver or adapter is not certain. The detector itself can take one of two forms. The first is the 'switching' type, complementary to the switching method of generating the complete modulation waveform. By gating parts of the discriminator waveform once every switching cycle in the appropriate phase, the L and R signals can be derived directly. The second form of detector is the phase sensitive detector which derives the component of the waveform in phase with the reference oscillator, and thus produces the difference signal.

A difficulty in all multiplex systems of the type being considered was mentioned in Section 5.4, namely that appreciable cross-talk between the L and R signals can result from quite small amplitude- or phase-response inequalities in circuits dealing with the sum and difference signals. Thus the mere absence of cross-talk between the sum and difference channels does not ensure freedom from cross-talk between the L and R channels. This has to be borne in mind in the design of receivers, and phase correction of the sum and difference audio-frequency signals may be needed. The switching technique and certain other methods of detection for the a.m. subcarrier systems have the advantage of avoiding phasing errors associated with the subcarrier demodulating circuits.

6.3.3 *Summarizing Remarks Concerning Multiplex Systems*

Some price must always be paid in terms of signal-to-noise ratio when using multiplex systems capable of giving two complete a.f. channels. In the case of a.m. transmissions this degradation need not be great providing some non-linear distortion is accepted, in monophonic reception at least, when the modulation depth is large. In the case of f.m. transmissions with a.m. or f.m. subcarrier, accurate prediction of the performance from theory is not possible because some effects, such as that of impulsive interference, are not amenable to calculation; experiments leading to the signal-to-noise deterioration obtained in practice and to the possible increase in adjacent- and co-channel interference are therefore necessary. For a monophonic service the performance of present receivers must be accepted, but for a stereophonic service, which requires new receiving equipment, any simple circuit refinements that can reduce the degradation in signal-to-noise ratio at the fringe of the service should be given due consideration.

In assessing any proposed system with given standards of modulation it is desirable, first for the monophonic service and then for the stereophonic service, to indicate the increase in field-strength necessary to give the same standard of reception as at present; some allowance for improved aerial installations and receiving equipment might be made in considering the stereophonic service but this is hardly permissible in the case of the monophonic service. From this information it would then be possible to derive the increases in the power or number of transmitters that might be required to maintain the existing or a slightly reduced standard of reception for either of the two types of service.

A further consideration not directly related to its performance on stereophonic transmissions might govern the choice of a multiplex system. It seems unlikely that all broadcast programme items would be suitable for stereophonic presentation and there may therefore be periods during which the full potentialities of a compatible transmission system are not realized. On such occasions it might be possible to provide an alternative monophonic programme on the channel normally reserved for the difference signal, but to do this requires a lower degree of cross-talk between the sum and difference channels than application to stereophony alone would dictate. However, this possibility is envisaged by the Nederlandse Radio-Unie,[23,24] and may well influence the choice of system in some countries. Reception of the second programme would, of course, require the receiver for stereophonic broadcasts to have an extra switched position for listening to the subcarrier signal alone, and moreover would be subject to some 3 dB greater degradation in signal-to-noise ratio than the stereophonic service, as indicated in Tables 2 and 3 for the $\frac{1}{2}(L-R)$ channel. Nevertheless, for a purely local service this effect may be unimportant, and with certain types of programme, such as news bulletins and background music, for which the requirements regarding the level of cross-talk are not very severe, the f.m. subcarrier system appears to offer at least one method which would give acceptable results.

6.4 *Systems using a Single Channel with Coding of the Directional Information*

A system being developed by Percival[28] at E.M.I. differs from the others so far described in adopting a fundamental approach to the problem of reducing to a minimum the information to be transmitted. The method of operation, reduced to its simplest elements, is to transmit a single programme signal which at the receiving end is divided between two loudspeakers in a ratio controlled by an auxiliary narrow-band 'steering' signal. This steering signal, which depends only on the angle of incidence of the sound arriving at the microphones, may be transmitted, in the case of a v.h.f./f.m. service, as a modulated subcarrier having a frequency of the order of 20 kc/s, while the programme signal is transmitted in the normal way and can be reproduced monophonically by existing receivers.

The idea of using a steering signal to operate upon a single programme channel in this way is quite old. The novelty of the present system lies in its application to a situation where a number of sound sources, at different angles with respect to the microphones, have to be simultaneously transmitted. Simultaneous presentation of a number of separated sound images would at first sight appear to be impossible since the ratio of the signals applied to the two receiving loudspeakers—on which ratio (other things being equal) the position of the image depends—can have only one value at one time. Recent hearing studies by Percival have, however, shown that the ap-

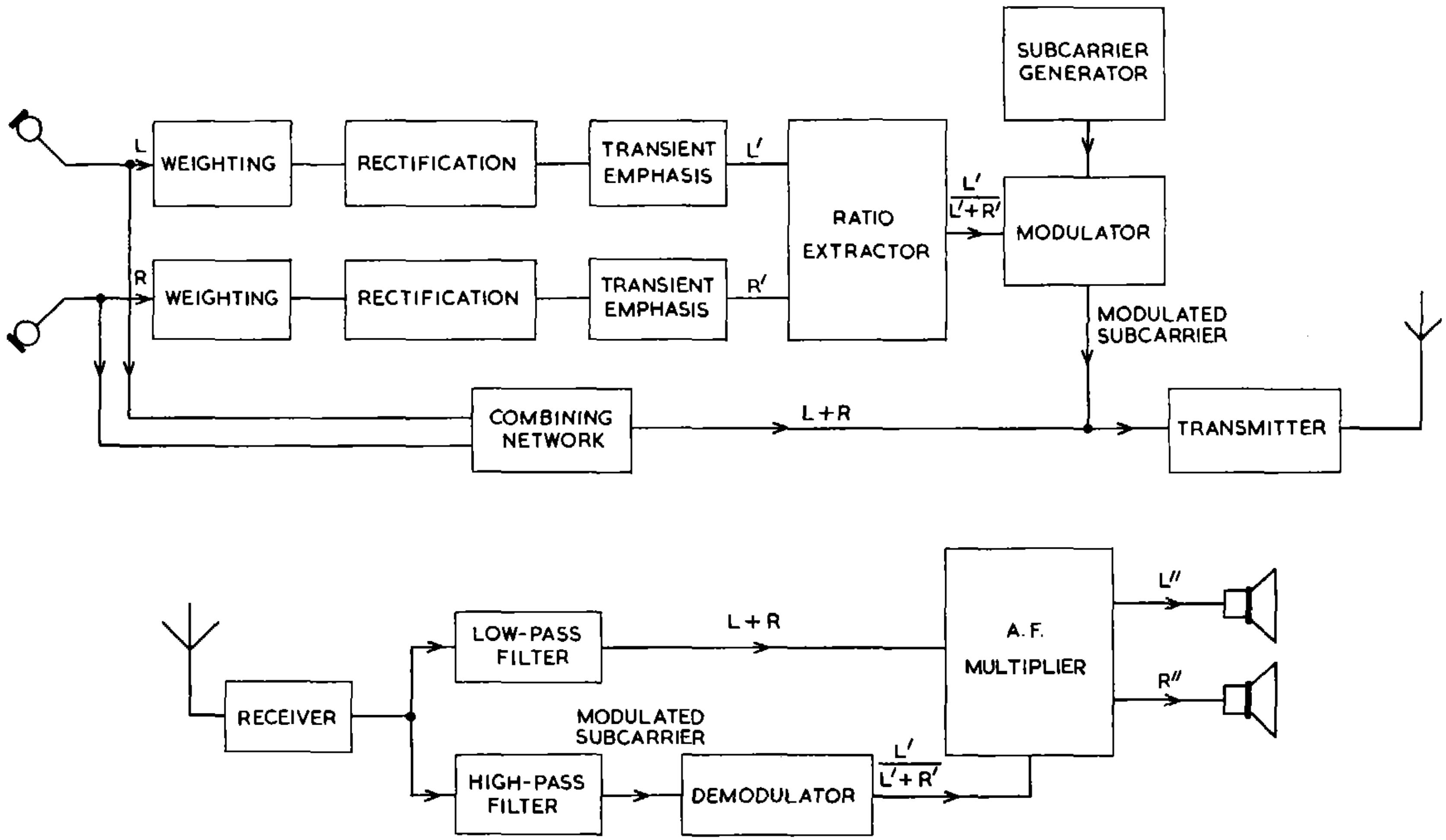

Fig. 6 — Percival system

parent position of a stereophonic image is largely decided at the moment of onset of the sound from the loudspeakers and that this subjective impression of position, once established, can persist for a short time even though the ratio of the loudspeaker outputs is subsequently altered. If, during this time interval, sound from a second source is transmitted, the apparent position of its image will likewise be determined by the value of the steering signal at the moment of onset, without however affecting the subjective impression of position already produced by the first source. As most programme material consists of sounds which are constantly starting and stopping or at least fluctuating in amplitude, the process can be made almost continuous, the positions of the various sound images being established in rapid succession and re-established at slightly longer intervals before the earlier impression has had time to wear off. It has been found possible to produce the illusion of sounds apparently coming from different directions at the same time when the bandwidth occupied by the steering signal is as low as 10 c/s, which corresponds to the transmission of twenty different directional 'instructions' per second. However, to permit a rapid change in direction between one image and the next, a bandwidth of about 100 c/s is in practice allowed.

The system requires for its operation a pair of coincident directional microphones, yielding signals L and R which for any one sound source differ in amplitude but are otherwise identical. The ratio of the two signal amplitudes depends on the direction of sound incidence and it is this ratio therefore which is used as a basis for the auxiliary steering signal. Fig. 6 shows in schematic form the coding and decoding processes involved. To extract the required directional information, it has been found necessary as the first operation to weight each of the incoming programme signals L and R by a network having a response approximately proportional to frequency. The resulting signals are rectified and each envelope is then operated on by a 'transient-emphasis' circuit designed to discriminate in favour of signals having a high rate of rise, thus giving two processed signals L′ and R′.

It remains now to derive some quantity dependent on the ratio of L′ to R′ but independent of the magnitude of these signals. This end is achieved by combining L′ and R′ in a series of operations, involving a temporary conversion to logarithmic units, yielding a steering signal

$$\frac{L'}{L'+R'}$$

which can vary between zero for a sound source on the extreme right and unity for a source on the extreme left. Finally, a subcarrier, generated by a local oscillator, is modulated by the steering signal and is fed, together with the combined audio-frequency programme signal (L+R) from the two microphones, to the input of a f.m. transmitter.

At the receiver, the subcarrier is extracted after the detector by a band-pass filter and demodulated to yield the steering signal

$$\frac{L'}{L'+R'}$$

This in turn is applied to an a.f. multiplier circuit arranged to give two output signals having amplitudes respectively

$$\frac{L'}{L'+R'} \text{ and } \left(1-\frac{L'}{L'+R'}\right)$$

times the audio-frequency signal L+R from the detector; in this way, signals

$$L''=\frac{L'}{L'+R'}(L+R)$$

and $$R''=\left(1-\frac{L'}{L'+R'}\right)(L+R)=\frac{R'}{L'+R'}(L+R)$$

are obtained. These derived signals, which are in the same ratio to one another as the original L and R signals, are amplified and fed to the two loudspeakers. Two points about the 'decoding' operation at the receiver are of special technical interest. The first of these is the use of a linear multiplier based on the Hall effect; with this arrangement, the usual difficulty of keeping the controlling signal from appearing in the programme circuit is completely absent. The second is the possibility, by some elaboration of the receiver circuits, of using a third loudspeaker, centrally placed; it would then be possible to reduce considerably the displacement of the sound image with movement of the observer.

Because the amount of directional information to be transmitted in the Percival system is small, it is possible to provide both monophonic and stereophonic services without any appreciable sacrifice in signal-to-noise ratio. In addition, the system might be expected not to place such stringent requirements on the performance of the programme lines to the transmitters as do the multiplex systems previously considered. The system also allows considerable latitude in methods of deriving the monophonic programme signal. Thus, the left- and right-hand signals can be so phase-shifted before being added as to bring corresponding frequency components into phase quadrature; by this device, the difficulty referred to in Section 4.3 of producing a suitable resultant directional characteristic for the microphones is lessened. Alternatively, provided that the steering signal is derived from a single pair of coincident directional microphones it should be possible to employ one or more separate microphones to produce the programme signals; this method of operation would, however, require three instead of two programme channels between the studio and the coding equipment. Nevertheless, the reservations already made regarding the compatibility of monophonic and stereophonic programmes derived from the same set of microphones apply also to the Percival system.

At the time of writing the Percival system is still in the experimental stage. Some of the initial difficulties have been cleared up and research at E.M.I. is continuing in order to clarify the situation, so that an overall assessment of the system can be made for broadcasting.

Some work on similar lines, carried out in Germany, has been described by Enkel.[29] Two or more pilot signals are transmitted at low level in the frequency range 14 kc/s to 15 kc/s, just above the upper frequency limit of the pro·

gramme signal. At the receiver, the pilot signals are separated from the noise by narrow band-pass filters and used to control the input level to the reproducing loudspeakers. The work is in a relatively early stage of development but serves to indicate the variety of the artifices which may be employed in this form of approach.

7. Application of Stereophony to Television

This monograph would not be complete without some reference to the potentialities of stereophonic sound transmission as an adjunct to a television picture. There is at present little or no information in the literature which could be directly applied to domestic reception conditions, though some three-channel stereophonic experiments, involving the use of two sound receivers tuned to different transmissions and placed on either side of the television receiver, have taken place in the U.S. The application of stereophony to television has recently been accepted as a question by the C.C.I.R.

It seems probable that stereophonic sound, properly presented by loudspeakers disposed symmetrically to left and right of the screen, could considerably enhance the interest and realism of a television programme, but a difficult situation might arise if the observer were thereby presented with directional information which conflicted with that provided by the picture. Experience in the cinema suggests that in the absence of any strong influence to the contrary, the eye is likely to take precedence over the ear in deciding the point from which sound appears to emanate, but the more effective the stereophonic system, the more important would it be to avoid at least the grosser incongruities between sound and vision. With the usual spacing of 5 to 10 feet between the left- and right-hand loudspeakers, the sound stage would be too wide for the size of picture provided by most domestic equipment. This disparity could be corrected by reducing the distance between the two loudspeakers, or by the roughly equivalent electrical device of introducing some degree of cross-mixing between the left- and right-hand channels, but much of the spacious effect which is one of the most attractive features of stereophony would then be lost. It might be possible, however, by a suitable combination of microphones, with the addition, if necessary, of artificial reverberation, to transmit the direct sound on a relatively narrow front while allowing the reverberant sound together with any 'noises off', such as applause, to extend over the maximum width.

In all but the simplest forms of presentation, the stereophonic reproduction accompanying a picture might well have to follow such cinematic practices as panning across a scene and frequent switching from one viewpoint to another. In this case, the resulting programme would be unacceptable for a sound broadcast; the sharing of a stereophonic programme signal with the sound service might therefore prevent full exploitation of the combined media.

8. Conclusions

The addition of a stereophonic service to an existing broadcasting system is clearly feasible as an engineering proposition, but poses a number of serious economic problems. To provide compatible stereophonic reception would probably require signal processing or coding equipment at studio centres or transmitters. Most of the compatible systems proposed would inevitably necessitate either an increase in the number or power of transmitters, a reduction in the service area, or the acceptance of a lower standard of reception; the others so far proposed involve instead some deterioration in the stereophonic effect. In addition, a radical change in the present system of programme distribution by line would be required. Whatever measures are to be adopted must depend for their justification on the degree of public demand for the benefits of stereophony and the proportion of listeners who are prepared to equip themselves, if necessary, with special receivers or adapters. For a realistic approach to the situation, it is necessary to consider what proportion of broadcast programmes would derive substantial benefit from stereophonic presentation, and to this end it will be necessary to gain experience with all types of programme. For cases in which stereophonic presentation is considered impracticable or pointless, it will be necessary to decide on the best method of transmission. Some time must elapse before the picture becomes clear, but if a demand for a stereophonic service develops it is likely to have an appreciable influence on the future of sound broadcasting.

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